(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
6 November 2003 (06.11.2003)

PCT

(10) International Publication Number
WO 03/092170 A1

(51) International Patent Classification⁷: H03M 13/45, 13/29, H04L 1/00

(21) International Application Number: PCT/NO03/00129

(22) International Filing Date: 22 April 2003 (22.04.2003)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
20022004 26 April 2002 (26.04.2002) NO

(71) Applicant *(for all designated States except US)*: **KONGSBERG DEFENCE COMMUNICATIONS AS** [NO/NO]; Postboks 87, N-1375 Billingstad (NO).

(72) Inventors; and

(75) Inventors/Applicants *(for US only)*: **OTNES, Roald** [NO/NO]; Engelsvikenveien 48D, N-1628 Engalsvik (NO). **TÜCHLER, Michael** [DE/DE]; H. Dransfeld-Allee 38, 80638 München (DE).

(74) Agent: **OSLO PATENTKONTOR AS**; Postboks 7007M, N-0306 Oslo (NO).

(81) Designated States *(national)*: AE, AG, AL, AM, AT (utility model), AT, AU, AZ, BA, BB, BG, BR, BY, BZ, CA, CH, CN, CO, CR, CU, CZ (utility model), CZ, DE (utility model), DE, DK (utility model), DK, DM, DZ, EC, EE (utility model), EE, ES, FI (utility model), FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, MZ, NI, NO, NZ, OM, PH, PL, PT, RO, RU, SC, SD, SE, SG, SK (utility model), SK, SL, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, YU, ZA, ZM, ZW.

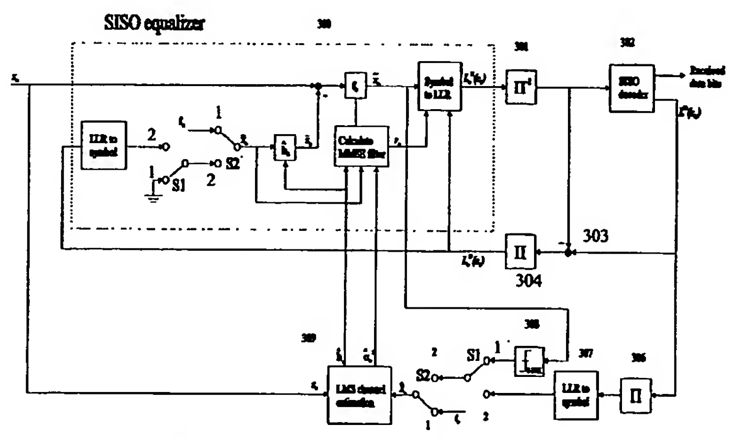
(84) Designated States *(regional)*: ARIPO patent (GH, GM, KE, LS, MW, MZ, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HU, IE, IT, LU, MC, NL, PT, RO, SE, SI, SK, TR), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

Declarations under Rule 4.17:

— *as to applicant's entitlement to apply for and be granted a patent (Rule 4.17(ii)) for the following designations AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EC, EE, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, MZ, NI, NO, NZ, OM, PH, PL, PT, RO, RU, SC, SD, SE, SG, SK, SL, TJ, TM, TN, TR, TT, TZ, UA, UG,*

[Continued on next page]

(54) Title: METHOD AND APPARATUS FOR THE RECEPTION OF DIGITAL COMMUNICATION SIGNALS



The two switches denoted S1 shall be in the position shown during first-time equalization, and in the opposite position during later iterations
The two switches denoted S2 shall be in the position shown during known symbols, and in the opposite position during data symbols

(57) Abstract: An apparatus and method for the detection of a coded digital signal received through a communication channel, in particular for the reception of serial-tone waveforms for High Frequency communication, involving a Soft In Soft Out (SISO) equalizer (300) and a SISO decoder (302) forming an iterative Turbo equalizer loop, a channel estimator (309) supplying channel impulse response estimates to the SISO equalizer (300), a source of known symbols (t_n), first switching means (S2) adapted to supply either intrinsic and extrinsic soft symbols from the SISO decoder (302) to the channel estimator (309) when receiving symbols carrying information, or supply known symbols (t_n) to the channel estimator (309) and the SISO equalizer (300) when receiving training sequences, and a hard decision mapper (308) receiving soft symbols from the SISO equalizer (300) and second switching means (S2) adapted to supply hard decision symbols to the channel estimator (309) from said mapper (308).

WO 03/092170 A1

UZ, VC, VN, YU, ZA, ZM, ZW, ARIPO patent (GH, GM, KE, LS, MW, MZ, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HU, IE, IT, LU, MC, NL, PT, RO, SE, SI, SK, TR), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG)

— *of inventorship (Rule 4.17(iv)) for US only*

Published:

— *with international search report*

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

A METHOD AND APPARATUS FOR THE RECEPTION OF DIGITAL COMMUNICATION SIGNALS

Field of the invention

The present invention relates to the reception of digital communication signals transmitted over an air interface, by using iterative channel estimation, and iterative equalization and decoding, also known as Turbo equalization. It is particularly applicable for baseband processing in a receiver in a High Frequency (HF) modem. The following description will focus on such an application. However, the inventive solutions may also find applications in other systems relying on wireless transmission links, e.g., in mobile communication systems.

Technical background

Communication in the HF bands (3-30 MHz) offers one of the few alternatives for beyond line-of-sight communication without relying on existing infrastructure or satellites. However, there are physical and regulatory limitations on the data rates achievable in HF systems. As to the last factor; the bandwidth is limited to 3 kHz due to frequency allocations. Most HF systems rely on signals reflected from the ionosphere, which varies with the time of day and year and the sun spot cycle. To a certain extent, these variations can be predicted, but the ionosphere also degrades the signals making high-rate digital communications difficult. This is especially the case at high latitude paths because of auroral effects. Rapid fluctuations in the ionosphere give rise to Doppler spread (fading), multiple propagation paths lead to delay spread, also known as multipath, which introduces intersymbol interference, and auroral absorption reduces the received signal–to-noise ratio.

In this invention we are primarily dealing with inter-symbol interference (ISI). In a signal experiencing such interference, adjacent symbols are interfering with each other, causing difficulties for the receiver trying to detect the individual symbols and read the information content.

The communication modes concerned are Phase Shift Keying modulation (PSK) or Quadrature Amplitude Modulation (QAM). In particular reference is made to the standards STANAG 4285, STANAG 4539 and MIL-STD-188-110B.

A number of techniques are introduced in order to improve the reliability of the data transfer. These include error correction coding, interleaving, and multiplexing of training sequences into the transmitted data stream.

On the receiving end equalization is used to combat the defects in the transmission channel. Currently, standard receivers apply decision feedback equalizers using non-linear processing for this task. A channel estimation process is used to estimate the channel impulse response, which is used to calculate the coefficients of the equalizer. The channel estimation process uses known symbols transmitted over the channel, i.e., the training sequence, to estimate the channel impulse response. Between the training sequences, tentative decisions from the equalizer can be used to track the time-varying channel. Alternatively, the channel impulse response between training sequences can be estimated through interpolation.

In recent years iterative equalization and decoding, or Turbo equalization, has been used with advantage as a powerful approach for receiving ISI-disturbed data. In a receiver applying Turbo equalization, the received signal is first passed through an equalizer in order to reduce ISI. A deinterleaver and a channel decoder is following the equalizer. The decoder outputs soft information on the code bits, which is fed back through an interleaver to the equalizer. Then, another equalization task is performed on the same received signal, using the soft bits fed back from the decoder as *a priori* information. When soft information is iterated between the equalizer and the decoder several times, the error rate of the data bits provided by the decoder will generally decrease.

In a receiver applying Turbo equalization, the equalizer and decoder are soft-in soft-out (SISO) modules. The optimal SISO equalizer is the trellis-based maximum *a posteriori* probability (MAP) equalizer, which is too complex for HF modems when the channel impulse response is long and/or the signal constellation is larger than 2-PSK. Suboptimal SISO equalizers based on linear filters and soft ISI cancellation are a good alternative when MAP equalization is too complex (M. Tüchler et al.: "Minimum Mean Squared Error Equalization Using A-priori Information", IEEE Trans. Sig. Proc., vol. 50, no. 3, pp. 673-683, Mar. 2002).

Prior art

The use of Turbo equalization in a receiver for digital communication signals was first described in C. Douillard et al.:"Iterative Correction of Intersymbol Interference: Turbo-Equalization", European Trans. Telecommun., vol. 6, no. 5, pp. 507-511, Sept.-Oct.

1995. This approach uses a trellis-based equalizer, which is too complex to be used for the application considered here. A reference discussing the relative merits of various trellis-based equalizers can be found in G. Bauch & V. Franz: "A comparison of soft-in/soft-out algorithms for Turbo-detection", Proc. Int. Conf. on Telecommunications, pp. 259-263, June 1998.

Later solutions have opted for a suboptimal approach using SISO equalizers based on linear filters, e.g., as described in A. Glavieux et al.:"Turbo equalization over a frequency selective channel," Proc. Int. Symp. on Turbo codes, Brest, France, pp. 96-102, Sept. 1997.

The papers mentioned so far are related to Turbo equalization, i.e., the problem of channel estimation in order to find the optimal coefficients for the equalizer is not considered.

In N. Nefedov & M. Pukkila:"Turbo Equalization and Iterative (Turbo) Estimation Techniques for Packet Data Transmission," 2nd Int. Symp. on Turbo Codes & Related Topics, Brest, France, pp. 423-426, 2000 is described an approach involving both a Turbo equalization loop and a separate loop for the estimation of the channel impulse response. The later loop includes a channel estimator receiving hard decisions from the decoder. Initially, the parameters are estimated from a training sequence, but later on, the estimate is updated from information symbols. However, this is a stepwise process, where the parameters are updated at intervals and locked to the current values during a complete frame.

From P. Strauch et al.: "Iterative Channel Estimation for EGPRS," IEEE52nd Vehicular Tech. Conf., Boston, USA, vol. 5, pp. 2271-2277, Sept. 2000 is known a conventional (non-Turbo) receiver in which the channel estimate is formed based on hard feedback either from the detector or the decoder during training sequences. The authors point to the possibility of using soft feedback for extending the training sequences.

In the patent literature are a number of approaches describing the use of a basic Turbo feedback loop for detection of communication signals in cellular communication systems, e.g., GB-application 2,354,676, EP-application 959,580 and EP-application 948,140. However, none of these documents considers the question of how the equalizer parameters should be estimated.

Brief summary of the invention

It is an object of the present invention to provide a method and an arrangement in a receiver with Turbo equalization for the reception of digital communication signals, with an improved ability of handling inter-symbol interference.

This is achieved in an apparatus according to the appended claim 1 and a method according to claim 8. The invention is applicable in a receiver using a SISO equalizer based on soft ISI cancellation and a linear filter. According to the inventive method, the coefficients of the SISO equalizer are updated at every symbol interval using a time-varying channel estimate and the *a priori* information on the code bits.

The scope of the invention will appear from the appended patent claims.

Brief description of the drawings

The invention will now be described in reference to the appended drawings, in which:

Fig. 1 shows a schematic representation of the encoding process used for PSK data transmissions,

Fig. 2 is a block diagram showing the principle of Turbo equalization,

Fig. 3 shows a corresponding receiver using Turbo equalization according to the present invention,

Fig. 4 is a graph showing the improvement in signal-to-noise-ratio that is achievable with the present invention.

Detailed description of the invention

Fig. 1 is a schematic representation showing the coding stages of a transmitter encoding an incoming data stream into a PSK signal.

The incoming data bits are encoded using a code with error-correction capability, in a coder 1. The code has the ability to correct errors occurring in individual symbols spread out at random. However, interference will often appear as burst noise (e.g. static energy), which will affect several adjacent symbols. Fading will also cause several

adjacent symbols to have low received signal level. Bursty error patterns are not easily handled by the error correcting code. To avoid this problem the signal is interleaved in the interleaver 2.

In the next step the data bits are modulated into data symbols in the symbol mapper 3. Thereafter training sequences containing known symbols are multiplexed into the data stream at 4. The training sequences are repeated in each block of data frames.

The stream of modulated data symbols are handled further by the transmitter circuitry, delivered to an antenna and transmitted on air. In the transmission channel noise is added and the symbols are distorted due to multi-path fading. The fading is a frequency selective time-varying phenomenon, or expressed in another way, the channel's impulse response is constantly changing.

The following description assumes that the received baseband signal is sampled once per symbol interval. The invention can also be modified as to use several samples per symbol, so-called fractionally spaced equalization or oversampling.

Fig. 2 shows the general principle of a receiver with Turbo equalization. Symbols obtained from a radio receiver (not shown) are fed to a Soft In Soft Out (SISO) equalizer 200. The soft information at the input and output of a SISO module takes the form of Logarithmic Likelihood Ratios (LLRs) on the code bits, $L(c_i)$:

$$L(c_i) = \ln\frac{\Pr(c_i = +1)}{\Pr(c_i = -1)}$$

Further information on Logarithmic Likelihood Ratios is available in: J. Hagenauer et al.: "Iterative decoding of binary block and convolutional codes," IEEE Trans. on Information Theory, pp. 429-445, March 1996.

The LLRs out of the equalizer are deinterleaved at 201, and decoded in the SISO decoder 202 (for the error correction code). The LLR values from the decoder are obtained as intrinsic information, i.e. the information content in the input soft information, and extrinsic information, which is information generated from the redundancy in the code. The extrinsic information is obtained by subtracting the LLRs at the input of the decoder from the corresponding LLR values at the output of the decoder in the subtractor 203. The resulting LLRs are fed back to the equalizer 200 through an interleaver 204. The interleaver 204 is included in order to produce signals that can be compared with the interleaved signals in the equalizer 200 (the equalizer 200

being upstream of the deinterleaver 201). The feedback information is used as *a priori* information in a new equalization and decoding attempt on the same set of data – being the first iteration. After a few iterations the error rate improves significantly. In this manner equalization and decoding are performed repeatedly for a fixed number of iterations, or until a termination criterion stops the iterative (Turbo) process.

The Turbo process described so far is dependent on the channel impulse response being known to the equalizer. A separate channel estimation process (not shown in Fig. 2) is therefore needed in order to provide an estimate of the channel impulse response.

Fig. 3 shows a receiver with Turbo equalization according to the present invention. The Turbo loop comprises a SISO equalizer 300, a deinterleaver 301, a SISO decoder 302, a subtractor 303 and an interleaver 304.

The SISO equalizer used here is based on soft inter-symbol interference cancellation followed by a linear filter. The LLRs fed back from the decoder are converted to an *a priori* expectation (mean) of the transmitted symbols, from which an expectation of the inter-symbol interference can be calculated. Soft inter-symbol interference cancellation subtracts this expected value of the inter-symbol interference from the received signal. The coefficients of the following linear equalizer filter are calculated as to minimise the expectation of the squared error (minimum mean square error criterion, MMSE) of each symbol output from the equalizer, given the *a priori* information and the estimated channel impulse response. The inner workings of the SISO equalizer are described in detail in Tüchler 2002, but the method has been modified as to incorporate a time-varying channel estimate. The new method is described in the Appendix.

Extrinsic LLRs for each code bit are calculated from the output symbols of the equalizer using the formula in Tüchler 2002.

Other SISO equalizers can also be used within the scope of this invention.

In addition to the turbo feedback loop, an additional channel estimator loop has been introduced. This outer loop includes another interleaver 306 connected directly to the SISO decoder 302. This will produce the total LLR output from the decoder, including the intrinsic information. These LLRs are converted to soft symbols (the expected value of each symbol given the LLRs) using the formula in Tüchler 2002. In addition a demapper 308 is introduced, giving a hard output signal from the equalizer. Further, the signals are routed by means of two switches S1 and S2.

The channel estimator uses the transmitted and received symbols to provide a time-varying estimate of the channel impulse response. The transmitted symbols are known during training sequences. Outside the training sequences, estimates of the unknown information symbols must be used. For initial equalization, hard symbols directly from the equalizer output are used as estimates (in this case, the channel estimate has to be delayed). In subsequent iterations, soft symbols calculated from the decoder output are used. The switch S2 is used to change between training symbols and estimated symbols, and the switch S1 is used to change between hard symbols from the equalizer and soft symbols from the decoder. The turbo loop is still used for detection of the information symbols. In this way, the channel estimate will also be improved for each iteration.

Thus, the invention involves a two stage channel estimation and equalization process comprising the following individual steps:

1. Initial equalization. S1 is placed in position 1 (as shown in Fig. 3). S2 is toggled between a first position (when receiving training sequences), in which known symbols are fed to the channel estimator and the equalizer, and a second position in which channel estimates are made using hard decisions received from the signal mapper 308. In this case, a delay (not shown) equal to the delay in the equalizer has to be applied to the channel estimate.

2. Iterative channel estimation, equalization, and decoding. Then switch S1 is moved to position 2. This will close both loops and establish a normal operation of the receiver. The upper loop will form a conventional iterative Turbo equalization loop decoding soft detected symbols. The lower loop will now continuously update the channel estimate based on extrinsic and intrinsic information received from the SISO decoder 303. S2 will toggle between position 1 when receiving training sequences and position 2 when receiving information symbols. In this case, the channel estimate needs not be delayed.

The channel estimator may use any algorithm for adaptive filtering or adaptive system identification, e.g., the stochastic gradient least mean square (SG-LMS) or recursive least squares (RLS) algorithm, as described in S. Haykin, Adaptive Filter Theory, 3rd Ed., Prentice Hall, 1996. The channel estimator provides the equalizer a time-varying estimate of the channel impulse response and of the error variance. The error variance is defined as the variance of the error (noise) signal e_n when the real channel impulse response has been replaced by the estimated channel impulse response.

$$e_n = z_n - \sum_{i=0}^{L-1} \hat{h}_{i,n} x_{n-i}$$

With notation as in the Appendix.

Fig. 4 displays simulation results showing Bit Error Rate versus Signal-to-Noise-Ratio in a receiver using a conventional (non-Turbo) receiver using a decision feedback equalizer, and a receiver according to the present invention. The simulation is performed using the 2400 bps HF waveform defined in MIL-STD-188-110B on a fading channel defined as mid-latitude disturbed conditions in ITU-R F.1487.

The graphs are shown for four different configurations of the Turbo-equalizer. Without iterations, the receiver will give an inferior SNR compared to a conventional receiver. This is as expected, due to the SISO equalizer being a linear filter, as opposed to the decision feedback equalizer in the conventional receiver. However, already after one iteration the SNR has improved markedly. Two iterations improves the SNR further, while the improvement from two to three iterations is small. Compared with a conventional receiver using a decision feedback equalizer, an improvement in SNR of about 2-3 dB is attainable.

Patent claims

1. An apparatus for the detection of a coded digital signal received through a communication channel including a Soft In Soft Out (SISO) equalizer (300) and a SISO decoder (302) forming an iterative Turbo equalizer loop,
characterized in that the apparatus further includes a channel estimator (309) supplying channel impulse response estimates to the SISO equalizer (300), a source of known symbols (t_n), first switching means (S2) adapted to supply either soft symbols calculated from extrinsic information from the SISO decoder (302) to the channel estimator (309) when receiving symbols carrying information, or supply known symbols (t_n) to the channel estimator (309) and the SISO equalizer (300) when receiving training sequences.

2. An apparatus as claimed in claim 1,
characterized in that said first switching means (S2) is adapted to supply soft symbols calculated from intrinsic in addition to extrinsic soft information from the SISO decoder (302) in one of its positions.

3. An apparatus as claimed in claim 1 or 2,
characterized in that the apparatus further includes a hard decision mapper (308) receiving soft symbols from the SISO equalizer (300) and second switching means (S2) adapted to supply hard decision symbols to the channel estimator (309) from said mapper (308).

4. An apparatus as claimed in claim 3,
characterized in that said SISO equalizer (300) includes a linear equalizer adapted to minimize the Mean Square Error (MMSE), when *a priori* information and a time-varying channel estimate is available.

5. An apparatus as claimed in claim 4,
characterized in that said channel estimator (309) is adapted to estimate the impulse response of the transmission channel using a Recursive Least Squares (RLS) algorithm.

6. An apparatus as claimed in claim 4,
characterized in that said channel estimator (309) is adapted to estimate the

impulse response of the transmission channel using a Least Mean Squares (LMS) algorithm.

7. An apparatus as claimed in any of the preceding claims, characterized in that said SISO equalizer (300) is adapted to sample the received signal several times per symbol interval.

8. A method for the detection of a coded digital signal received through a communication channel, which signal is equalized in a SISO linear equalizer, and decoded in a decoder, said equalizer and decoder forming an iterative Turbo equalizer loop,
characterized in that the impulse response of the communication channel is estimated in
a) an initial step in which estimates are formed from known training symbols when receiving training sequences multiplexed into the signal stream and from hard decisions received from the equalizer when receiving information symbols,
b) a subsequent step in which estimates are formed from known training symbols when receiving training sequences and soft symbols from the decoder when receiving information symbols.

9. A method as claimed in claim 8, characterized in that in step b) said soft symbols comprises extrinsic soft information from the decoder.

10. A method as claimed in claim 8, characterized in that in step b) said soft symbols comprises intrinsic and extrinsic soft information from the decoder.

11. A method as claimed in claim 9 or 10, characterized in that received symbols is equalized to minimize the Mean Square Error (MMSE), when *a priori* information and a time-varying channel estimate is available.

12. A method as claimed in claim 11, characterized in that the equalizer estimates each transmitted symbol as:

$$\hat{x}_n = \mathbf{f}_n^H (\mathbf{z}_n - \bar{\mathbf{z}}_n + \bar{\mathbf{x}}_n \hat{\mathbf{s}}_n)$$

in which $\hat{\mathbf{s}}_n$ is the (N_1+1)th column of $\hat{\mathbf{H}}_n$, $\hat{\mathbf{H}}_n$ being the time-varying Nx$(N+L-1)$ channel convolution matrix, a band-diagonal matrix where the rows contain the vectors $\hat{\mathbf{h}}^T_{n+N_1}...\hat{\mathbf{h}}^T_{n-N_2}$ and zeros, $\hat{\mathbf{h}}_n$ being the time varying estimate of the channel impulse response, and $\mathbf{f}_n$ contains the coefficients at time step n of the equalizer filter, of length $N=N_1+N_2+1$.

13 A method as claimed in claim 12,
c h a r a c t e r i z e d i n that the equalizer filter is calculated as:

$$\mathbf{g}_n = \left(\hat{\mathbf{H}}_n \mathbf{V}_n \hat{\mathbf{H}}_n^H + \Sigma_n\right)^{-1} \hat{\mathbf{s}}_n = \mathbf{U}_n \hat{\mathbf{s}}_n$$
$$\mathbf{f}_n = \mathbf{g}_n / \left[1 + (1 - v_n) \mathbf{g}_n^H \hat{\mathbf{s}}_n\right]$$

where Σ_n is a diagonal matrix with the elements $\sigma^2_{n+N_1}...\sigma^2_{n+N_2}$, the time-varying estimate of the error variance, and the matrix $\mathbf{U}_n$ is calculated recursively from $\mathbf{U}_{n-1}$ using the following algorithm:

$$a_n = \sigma^2_{n+N_1} + \sum_{i=0}^{L-1} \left|\hat{\mathbf{h}}_{i,n+N_1}\right|^2 v_{n+N_1-i}$$
$$\mathbf{a}_n = \left[\alpha_{1,n} \cdots \alpha_{N-1,n}\right]^T$$
$$\alpha_{j,n} = \sum_{i=0}^{L-1-j} \hat{h}_{i,n+N_1-j} \hat{h}^*_{i+j,n+N_1} v_{n+N_1-i-j}$$
$$\mathbf{A}_n^{-1} = \mathbf{D}_n - \mathbf{d}_n \mathbf{d}_n^H / d_n$$
$$\psi_n = \mathbf{A}_n^{-1} \mathbf{a}_n$$
$$c_n = 1/(a_n - \mathbf{a}_n^H \psi_n)$$
$$\mathbf{c}_n = -\psi_n c_n$$
$$\mathbf{C}_n = \mathbf{A}_n^{-1} + c_n \psi_n \psi_n^H$$

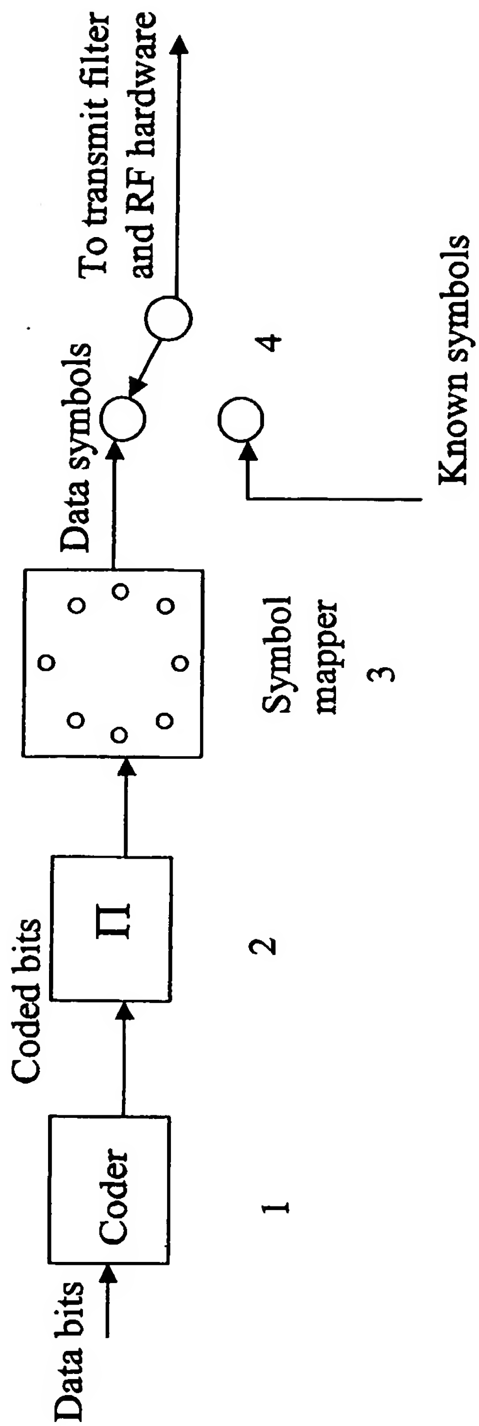
where $\mathbf{U}_{n-1}$ and $\mathbf{U}_n$ have been split as

$$\mathbf{U}_{n-1} = \left[\begin{array}{c|c} \mathbf{D}_n & \mathbf{d}_n \\ \hline \mathbf{d}_n^H & d_n \end{array}\right], \mathbf{U}_n = \left[\begin{array}{c|c} c_n & \mathbf{c}_n^H \\ \hline \mathbf{c}_n & \mathbf{C}_n \end{array}\right]$$

14. A method as claimed in claim 11 or 13,
c h a r a c t e r i z e d i n that said channel impulse response is estimated using a Recursive Least Squares (RLS) algorithm.

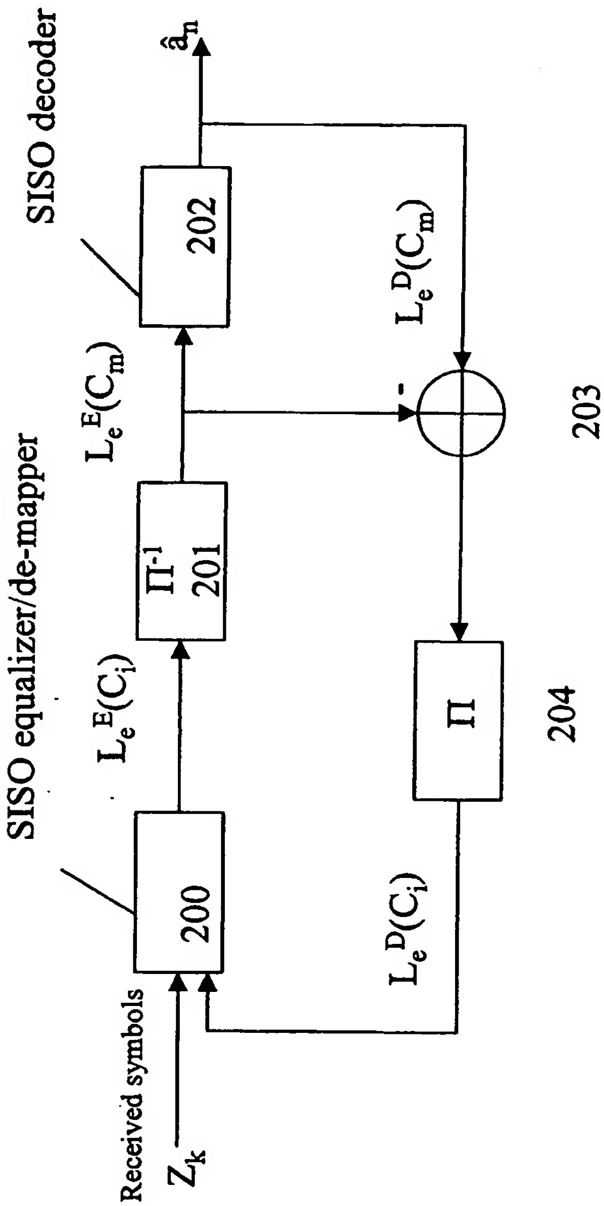
15. A method as claimed in claim 11 or 13,
c h a r a c t e r i z e d i n that said channel impulse response is estimated using a Least Mean Squares (LMS) algorithm.

16. A method as claimed in any of the claims 8-15,
c h a r a c t e r i z e d i n that the received signal is sampled several times per symbol interval.



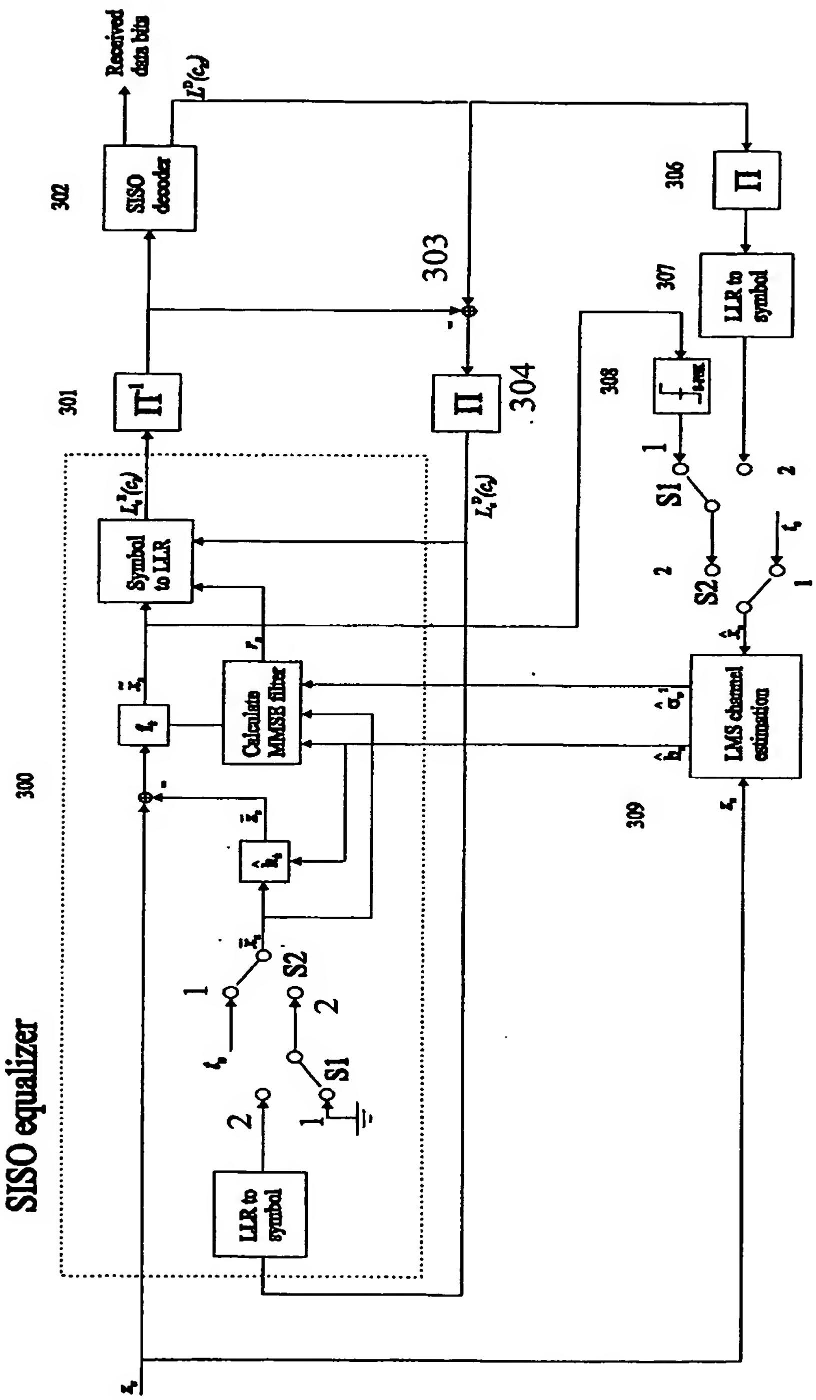
Data bits
Coder
1
Coded bits
Π
2
Symbol
mapper
3
Data symbols
4
Known symbols
To transmit filter
and RF hardware

Figure 1



SISO equalizer/de-mapper
SISO decoder
Received symbols
Z_k
200
$L_e^E(C_i)$
Π^{-1}
201
$L_e^E(C_m)$
202
$\hat{a}_n$
$L_e^D(C_m)$
-
203
Π
204
$L_e^D(C_i)$

Figure 2



The two switches denoted S1 shall be in the position shown during first-time equalization, and in the opposite position during later iterations
The two switches denoted S2 shall be in the position shown during known symbols, and in the opposite position during data symbols

Figure 3

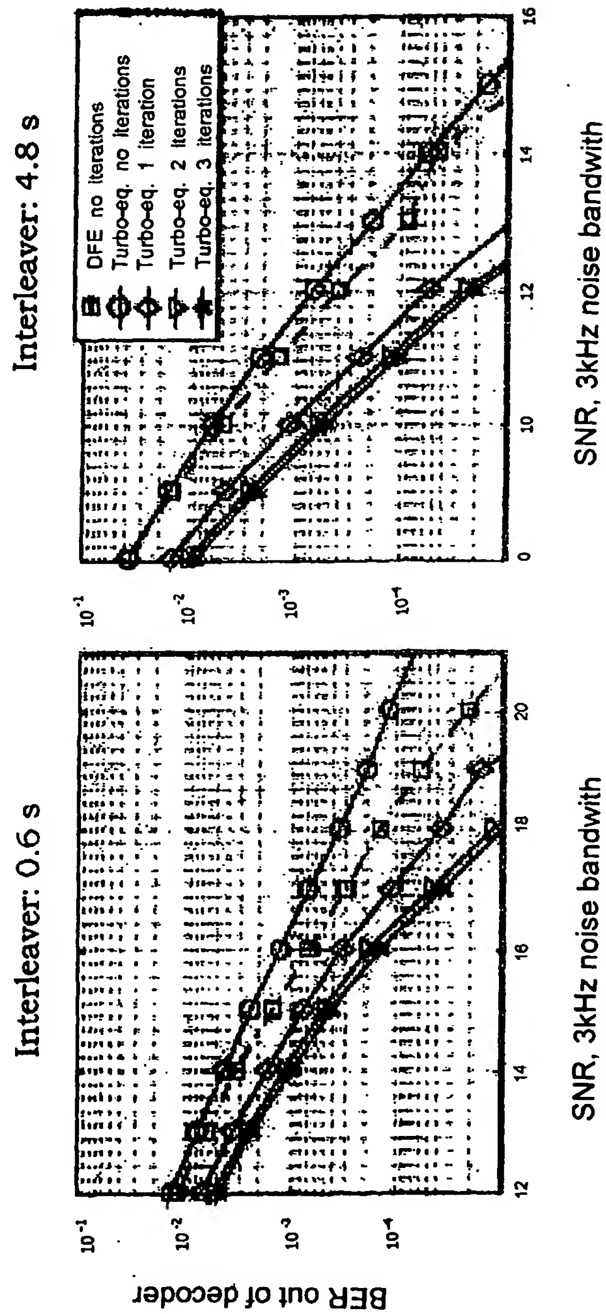

Interleaver: 0.6 s
Interleaver: 4.8 s
DFE no iterations
Turbo-eq. no iterations
Turbo-eq. 1 iteration
Turbo-eq. 2 iterations
Turbo-eq. 3 iterations
BER out of decoder
SNR, 3kHz noise bandwith
SNR, 3kHz noise bandwith

Figure 4

International application No.
PCT/NO 03/00129

A. CLASSIFICATION OF SUBJECT MATTER

IPC7: H03M 13/45, H03M 13/29, H04L 1/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC7: H03M, H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

EPO-INTERNAL, WPI DATA, PAJ, INTERNET, INSPEC

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
P,X	TUCHLER,M. ET AL.: Improved Receivers for Digital High Frequency Waveforms using Turbo Equalization. Proceedings from Military communications Conf. (MILCOM), Anaheim, USAOctober 2002. Retrieved from the Internet: http://www.lnt.e-technik.tu-muenchen.de/mitarbeiter/tuechler/tuechler.html See whole document --	1-16
A	EP 0954143 A1 (LUCENT TECHNOLOGIES INC), 3 November 1999 (03.11.99), figures 1,3-4, claims 1-7, abstract --	1-16
P,A	EP 1233565 A2 (NTT DOCOMO, INC), 21 August 2002 (21.08.02), claims 1,11, abstract -- --------	1-16

☐ Further documents are listed in the continuation of Box C. ☒ See patent family annex.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier application or patent but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance: the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance: the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
11 August 2003	13 -08- 2003
Name and mailing address of the ISA/ Swedish Patent Office Box 5055, S-102 42 STOCKHOLM Facsimile No. +46 8 666 02 86	Authorized officer Nabil Sebaa /LR Telephone No. +46 8 782 25 00

INTERNATIONAL SEARCH REPORT
Information on patent family members

02/06/03

International application No.
PCT/NO 03/00129

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
EP 0954143 A1	03/11/99	AU 726062 B	26/10/00
		AU 2401099 A	11/11/99
		BR 9901493 A	11/01/00
		CN 1236242 A	24/11/99
		JP 2000004190 A	07/01/00
		US 6459728 B	01/10/02
EP 1233565 A2	21/08/02	CN 1373561 A	09/10/02
		US 2002161560 A	31/10/02